Using an ACF message, MVAM or a third-party billing application, set the following timers:

Timer	Description
Call countdown timer	Sets the time remaining before a gateway disconnects the current call. When this timer expires, the gateway plays an announcement that time has expired and disconnects the call. • By default, once the timer on the gateway is set, the h323drq.au announcement file is played back for the caller upon call termination. • If the MVAM or third-party billing application uses its own countdown timer, the announcement specified in an Disengage Request (DRQ) message can be used to select a different announcement file for playback upon call termination.
Call disconnect warning timer	Specifies when a call disconnect warning announcement is played for the caller. This announcement alerts the caller to the time remaining before this call is terminated. • By default, once the timer on the gateway is set, the h323bkin.au announcement file is played back for the caller when the time expires. • If the MVAM or third-party billing application uses its own disconnect warning timer, the announcement specifier in an Interrupt Request (IRQ) massage can be used to select a different announcement file for playback when this timer expires.

H.323 call-specific administration messages

Call administration information is transmitted as part of the nonstandard data included in registration, admission and status (RAS) messages exchanged between the gateway and gatekeeper for each call. This data consists of a set of parameters using URL encoding, as described in RFC 1738, with each parameter composed of a set of attribute value pairs.

This nonstandard data may include the following call administration information:

- ANI/CLID
- Conference identifier
- User PIN
- Inbound or outbound trunk identification
- Enable voice announcement playback
- Select voice announcement playback
- Internal call timer and disconnect timer settings
- Call failures
- Call results

S 1469

- Trunk group and DS0 status information
- Available digital signal processors (DSPs)
- Maximum number of calls a MultiVoice Gateway may support

Trunk and call status reporting

Each MultiVoice Gateway reports its current call processing status as part of a Registration Request (RRQ) message to MVAM. The RRQ message includes data on trunk, trunk group, and DS0 status. The initial RRQ message, sent to MVAM when a gateway is initialized, contains a full report on all the trunks used by the physical gateway. The RRQ messages sent during keepalive registration include only the status changes since the previous registration message.

DS0 status (in-service/out-of-service)

A MultiVoice Gateway reports trunk, trunk group, and DS0 information to MVAM for each trunk. This includes:

- Trunk group
- Physical address
- DS0 service status (in-service or out-of-service)

Note A DS0 is in service for a logical gateway when it belongs to the associated trunk group and is in the up state. Information regarding DS0 activity (in use, free) is not reported via RRQ. DS0 activity is reported separately, traced from the per-call trunk/DS0 reporting mentioned below.

Trunk groups and physical address (shelf, slot, etc.) information are provided to MVAM to allow dynamic tracking of DS0 activity and trunk group assignments, and provided for future support of DS0 selection by physical-address for outbound PSTN calls.

Full trunk and DS0 status reporting is performed only when necessary, enhancing gateway performance. Full RRQs report complete trunk and DS0 information, usually when a gateway is initialized or else when requested by MVAM. Lightweight RRQs are used to report only status changes for trunk and DS0 information. MVAM can request complete trunk and DS0 information by responding to a lightweight RRQ with a Registration Reject (RRJ) message containing the reject reason `FullRegistrationRequired`.

Note Currently, trunk and DS0 status are not reported for BRI lines. Only the following information is reported for MultiVoice Gateways that use BRI:

- Number of idle VoIP ports.
- Value of `maxcalls` in `voip` profile.

Trunk and DS0 reporting (per call)

Trunk group and physical address information for the DS0 connection are reported for each call processed by a MultiVoice Gateway. This information is sent from the

gateway to the gatekeeper as nonstandard data in these registration, admission, and status (RAS) messages for the following call types:

Table 3-1. Trunk Messages

Message	Call type	Trunk or DS0 information
Admission Request (ARQ)	Inbound (from PSTN)	The trunk group and physical address of the DS0 upon which the call arrived.
Bandwidth Request (BRQ)	Outbound (to PSTN)	The trunk group and physical address of the DS0 upon which the call went out.
Disengage Request (DRQ)	Inbound (from PSTN) and Outbound (to PSTN)	The physical address of the DS0. **Note** For outgoing PSTN calls, the trunk group or DS0 information might not be present.
Disengage Confirmation (DCF)	Inbound (from PSTN) and Outbound (to PSTN)	The trunk group and DS0 information for gatekeeper-initiated call terminations.

Trunk and DS0 selection (per call)

Currently, MultiVoice Gateways only support trunk-group based routing for outbound PSTN calls. To do this, trunk groups must be enabled in the System profile of each gateway in the MultiVoice network. Each T1 or E1 line must also be assigned a trunk group.

Note Trunk groups should only be assigned at the T1 level.

The physical address information collected by the gateway for each DS0 is used currently by MVAM to track DS0 activity dynamically. The physical address is currently not used for DS0 to DS0 linking. In the future, both trunk group and/or physical address information will be available for DS0 selection on the gateway. When this happens, trunk groups should only be used when processing both VoIP and data calls on the same gateway. Otherwise, only gatekeeper, physical-address based, DS0 routing should be used.

gk-mlg-control parameter

The `gk-mlg-control` parameter in the `voip` profile enables the MultiVoice Gateway to accept and process call-specific administration instructions from MVAM 3.0.

When enabled, the gateway can apply call-specific processing instructions for PIN authentication, single- or two-stage dialing, voice announcement playback, and configuring call timers for prepaid billing. Values received from MVAM or third-party billing systems override parameter settings in the `voip` profile that process the current VoIP call.

S 1471

Rules used for performing call-specific administration are configured on MVAM and are used when partitioning MultiVoice Gateways into multiple logical gateways. This allows MVAM to administer a single physical gateway as if it were multiple gateways, partitioning the gateway according to trunk groups, DNIS, time of day, etc.

Call-specific administration is enabled by specifying yes, enabling the processing of call-specific administration instructions. The default, no, causes reversion to global administration of VoIP calls using the values set in the `voip` profile.

The following example illustrates how to enable multiple logical gateway processing on a TAOS unit:

```
admin> read voip { 0 0 }
VOIP/{ 0 0 } read

admin> set gk-mlg-control=yes

admin> write
VOIP/{ 0 0 } written
```

This parameter has the following dependencies:

- If `gk-mlg-control`=yes, the value of the `vpn-mode` parameter defaults to `N/A`
- If `gk-mlg-control`=yes, the value of the `single-dial-enable` parameter defaults to `N/A`

Changes to this parameter are effective with the next VoIP call.

Setting H.323 dialing options

MultiVoice offers a set of dialing options that support the following:

- Adjusting the amount of time a caller has to dial a telephone number
- Performing single-stage dialing of telephone numbers
- Rerouting blocked VoIP calls back out over the local PSTN
- Deactivate trunks used for VoIP calls
- Request operator assistance during the dialing phase
- Enable early ringback signaling
- Enable the use of trunk prefixes for routing VoIP calls out to the local PSTN
- Enabling user-entered authentication
- Determining the length of a dial string

Adjusting and troubleshooting the interdigit timer

The `call-inter-digit` parameter limits how long a TAOS unit waits for a caller to enter a single digit when using two-stage dialing. The customer may set the interdigit timer, for any value between 1 and 20 seconds, by changing the value of the `call-inter-digit-timeout` parameter.

By default, callers have 6 seconds to enter each digit of a telephone number, with a one-second decrement for each digit a caller enters. When the timer expires, the dialing is considered to be complete and the call proceeds. If callers finish dialing before the time expires, they can wait up to 16 seconds or press the pound (#) key before the gateway continues processing the call.

S 1472

Setting the `primary-retries` parameter in the `voip` profile to zero (0) disables this feature. You may enter any value between 300msec and 20,000msec (0.3 seconds and 20 seconds). Changes to this value become effective with the next registration cycle. This value defaults to 6000msec.

The following example illustrates how to set the value of the interdigit timer:

```
admin> read voip { 0 0 }
VOIP/{0 0} read
admin> list call-inter-digit-timeout
[in VOIP/{ 0 0 }:call-inter-digit-timeout]
call-inter-digit-timeout = 6000
admin> set call-inter-digit-timeout = 4000
admin> write
VOIP/{ 0 0 } written
```

The following dependencies apply to adjusting the inter-digit timer:

- This timer setting is applied to PIN entries and digits dialed after entering the telephone number.
- Values below 300 milliseconds may result in dropped digits.

Troubleshooting the interdigit timer

There are two common problems that result from setting the interdigit timer value too low. They are as follows:

Trouble	Corrective action
Enabling or resetting the interdigit timer may result in dropping dialed digits.	Dropping dialed digits usually occurs if the gateway is configured to wait for a short, 300msec to 1,000msec, time interval. To correct this problem, increase the time interval to 3000msec, or higher, depending upon the frequency and severity of the problem.
Enabling the inter digit timer caused single-stage dialing to fail	Under certain circumstances, enabling the interdigit timer can cause single-stage dialing to fail. When this occurs, try increasing the time interval for single-digit collection. If that fails to correct the problem, disable the configurable interdigit timer by turning digit collection off in the Line-Interface sub-profile of the T1 or E1 profile.

S 1473

Configuring single-stage dialing

The single-dial-enable parameter is used to enable or disable single-stage dialing of VoIP calls when MultiVoice is configured to perform H.323 call processing. You can enter either of the following values:

Parameter value	Specifies
yes	The TAOS unit extracts the Dialed Number Identification Service (DNIS) string for the destination telephone number from a single dialed entry. The destination number is passed to the distant gateway during call-setup.
no	(Default) That this feature is disabled. Callers are required to dial the TAOS unit, then wait for a subsequent dial tone before dialing the called telephone number.

The following example illustrates how to enable single-stage dialing on a TAOS unit:

```
admin> read voip { 0 0 }
VOIP/{ 0 0 } read
admin> set single-dial-enable = yes
admin> write
VOIP/{ 0 0 } written
```

Single-stage dialing works with MultiVoice Gateways under the following conditions:

- You are using T1 inband trunks, and the switch (or PBX) can relay DTMF signals to the MultiVoice Gateway.
- You are using T1 PRI trunks.
- You have enabled collection of DNIS/ANI on the TAOS unit.

For additional information see "Configuring trunk signaling for H.323 VoIP networks" on page 2-54.

Using H.323 single-stage dialing without PIN authentication

Users do not need to enter a Personal Identification Number (PIN) authentication to complete a VoIP call if vpn-mode = yes or users are authenticated using ANI. Callers enter only the MultiVoice access number followed by the destination phone number (DNIS). For example, they can enter 997325551212. The digits specify the following:

Table 3-2. Digits

99	The access number. This can be either single or multiple digits, configurable by the service provider. This number is not forwarded to the destination gateway.
7325551212	The destination phone number. This is a real destination number (DNIS) that must be sent to destination gateway. This number could be a PBX extension (such as 3103 in a company 's private phone network) or a full public phone number as shown here.

Using H.323 use single-stage dialing with PIN authentication

Users can enter the access number, followed by the destination phone number, and be prompted to enter their PIN to complete a VoIP call if `vpn-mode = no`. Callers enter the MultiVoice access number and destination phone number (DNIS) all at one time, then hear the PIN prompt (three short beeps). The user must enter the PIN to initiate call processing. In future releases, callers hear a voice announcement "Please enter you PIN number."

Rerouting blocked calls over the local PSTN

When a TAOS unit is unable to process an incoming voice call because registration with the gatekeeper fails, it can attempt to connect the call using its local PSTN connection.

This technique of turning the call back from the MultiVoice Gateway over the PSTN is called hairpin dialing. This allows a TAOS unit to complete calls over the public switched network when it is unable to route them over the IP network.

The `call-hairpin` parameter controls whether a TAOS unit will attempt to re-route blocked calls using its local PSTN connections. You may enter either of the following values:

Parameter value	Specifies
yes	The TAOS unit connects calls using the PSTN if it cannot register with a MultiVoice Gatekeeper (MVAM).
no	(Default) The TAOS unit does not connect calls using the PSTN if it cannot register with MVAM. New call requests are rejected until it successfully registers with a gatekeeper.

Changes to this value take effect with the next VoIP call.

To enable hairpin dialing on a TAOS unit for VoIP calls:

```
admin> read voip { 0 0 }
VOIP/{ 0 0 } read
admin> set call-hairpin = yes
admin> write
VOIP/{ 0 0 } written
```

Note Hairpin dialing only works when a second DSP is available in the same TAOS unit to handle the outbound call to the PSTN. That DSP may be on the same MultiDSP slot card or a second DSP slot card installed in the same shelf of the TAOS unit.

Requesting operator assistance

Callers can request operator assistance during the dialing phase of a MultiVoice call. A TAOS unit can be assigned a dial string, up to five-digits long, that can be entered by a caller to connect that caller to an operator.

Callers can enter a set of digits (such as: *0, 09, etc.) when they need operator assistance during the dialing stage of a MultiVoice call. The digit string used to request operator assistance is defined in the `operator-assist` parameter in the `voip` profile.

When the caller enters the operator assistance digits, the TAOS unit sends them to MVAM, which translates these digits into the actual number to dial for operator assistance. MVAM sends this number to the far-end MultiVoice Gateway to connect the call to an operator.

Once the call is connected, the digit string used to request operator assistance is available for normal call processing functions, such as responding to automated attendants, AUDIX, etc.

The operator assistance option is supported for MultiVoice Gateways operating as either multiple logical gateways (`gk-mlg-control` = yes) or as a single gateway (`gk-mlg-control` = no). To provide operator assistance requires MVAM 3.1.0 be installed and running on the gatekeeper.

operator-assist parameter

The `operator-assist` parameter defines the dial string a caller enters when requesting operator assistance. This parameter value can be up to five digits long.

The `operator-assist` feature is enabled by entering a two to five-digit dial string containing an asterisk (*) in either the first or second position. This parameter accepts the asterisk (*) plus any number(s) 0 through 9 as a valid entry. By default this value is *0. This feature is disabled by assigning a NULL value to the `operator-assist` parameter.

The following illustrates how to set the value of the `operator-assist` parameter:

```
tnt17>read voip { 0 0 }
VOIP/{ 0 0 } read

tnt17>set operator-assist = *9

tnt17>write
VOIP/{ 0 0 } written
```

To disable the operator assistance feature, set the value of the `operator-assist` parameter value as illustrated:

```
tnt17>set operator-assist =

tnt17>write
VOIP/{ 0 0 } written
```

The `operator-assist` parameter has the following dependencies:

- The first or second digit of the dial string must always be an asterisk (*).
- A MultiVoice Gateway must be configured for two-stage dialing (`single-dial-enable` = no).
- The gatekeeper must be running MVAM 3.1.0.
- A translation rule must be defined in one of the ingress translation tables used by MVAM that contains the actual dialed number used to connect calls to operator assistance.

Enabling early ringback

The `early-ringback-enable` parameter allows a TAOS unit to generate a ringback tone locally, as soon as the call is started on the far-end gateway. Early ringback eliminates delays in call notification times, which can occur in certain VoIP network configurations (such as satellite IP networks, wireless networks, or networks using

channel-associated signaling (CAS) trunks). Delays in call notification in these network environments can cause callers to hang up before the call completes, while waiting for call-progress tones from the far-end PSTN.

Caution Early ringback is intended for use only on networks that experience long call-setup times. Its use for other network configurations is not recommended and might result in erroneous ring-to-busy and ring-to-failure announcements.

The following settings enables early ringback:

```
admin> read voip { 0 0 }
VOIP/{ 0 0 } read
admin> set early-ringback-enable = yes
admin> write
VOIP/{ 0 0 } written
```

Enabling trunk prefixing

The trunk-prefix-enable parameter enables a TAOS unit to identify and assign an egress trunk group to the destination telephone number. When received by the egress MultiVoice Gateway or call signaling entity, the trunk group prefix is used to select the egress trunk to connect the call.

With trunk prefixing, the TAOS unit is able to identify the entry (ingress) trunk number to the exit (egress) gateway or call signaling entity by prepending the ingress trunk number to the DNIS number. Trunk groups must be in use system-wide.

When trunk prefixing is enabled, the system obtains the trunk group number from both:

- The trunk-group parameter in the T1 line profile associated with the inbound trunk on the ingress MultiVoice Gateway
- The ACF message from MVAM

Once assigned, the trunk group number is prepended to the destination telephone number. The trunk group/dial string combination is sent as the Q.931 Called Party Number information element (IE) in an H.225/Q.931 SETUP message to the egress MultiVoice Gateway. The destination address value of the SETUP user-to-user information element (UUIE) is not currently encoded.

Trunk-Prefix-Enable parameter

When set to yes, the trunk-prefix-enable parameter causes an egress MultiVoice Gateway to route outbound calls to the PSTN using a preselected trunk group, assigned by either the ingress MultiVoice Gateway or MAVM. When set to no, the default, the egress MultiVoice Gateway selects trunk groups for outbound calls.

For example, the following commands enable trunk prefixing, beginning with the next VoIP call the TAOS unit receives:

```
admin> read voip { 0 0 }
VOIP/{ 0 0 } read
admin> set trunk-prefix-enable = yes
admin> write
VOIP/{ 0 0 } written
```

This parameter has the following dependencies:

S 1477

- Using trunk groups must be enabled in the `system` profile on the egress MultiVoice Gateway (`use-trunk-groups` = yes).
- The size of the trunk groups must be defined (`num-digits-trunk-groups` = 1) in the `system` profile on all egress MultiVoice Gateways.
- Trunk group numbers must be assigned in both the T1 trunk and line profiles for egress T1 trunks.

Configuring PIN collection

The `vpn-mode` parameter enables or disables collection of a MultiVoice user's PIN by this TAOS unit when MultiVoice is configured to perform H.323 call processing. This parameter controls whether a user must enter a separate PIN code when placing a VoIP call.

User PINs are assigned by MVAM, whenever a new user is added to the gatekeeper's database. After a user enters a PIN, it is sent to the gatekeeper as part of the call admissions request (ARQ) from the TAOS unit. The gatekeeper then authenticates the user before continuing with call-setup.

You may enter either of the following values the `vpn-mode` parameter:

Parameter value	Specifies
yes	The TAOS unit does not prompt for a user-entered PIN. All calls are admitted without requiring user-entered authentication, as if the call were made on a virtual private network.
no	(Default) The TAOS unit prompts callers for their PINs before admitting calls. The TAOS unit presents callers with either a dial tone or prompts indicating that a user-entered PIN is required.

Note This parameter has no effect on performing Automatic Number Identification (ANI) authentication for H.323 call processing.

The following example illustrates how to disable user-entered PIN collection on a TAOS unit:

```
admin> read voip { 0 0 }
VOIP/{ 0 0 } read
admin> set vpn-mode = yes
admin> write
VOIP/{ 0 0 } written
```

Enabling sequential calls for PIN authentication

Callers who must enter a PIN to authenticate MultiVoice calls can dial subsequent VoIP calls without reentering their PINs, as long as they do not terminate the connection between the PSTN and near-end MultiVoice Gateway. MultiVoice users need only authenticate once, for the initial VoIP call, to initiate many subsequent calls.

Dialing the next call without authentication is supported for MultiVoice Gateways operating as either multiple logical gateways (`gk-mlg-control` = yes) or as single gateways (`gk-mlg-control` = no).

sequential-call-enable parameter

To enable the `sequential-call-enable` parameter, set the value to yes, the default. To disable the feature, set the value of the `sequential-call-enable` parameter to `No`.

The following procedure illustrates how to set the value of the `sequential-call-enable` parameter:

```
tnt17>read voip { 0 0 }
VOIP/{ 0 0 } read

tnt17>set sequential-call-enable = yes

tnt17>write
VOIP/{ 0 0 } written
```

To disable the sequential call dialing feature, set the value of the `sequential-call-enable` parameter as illustrated:

```
tnt17>set sequential-call-enable = no

tnt17>write
VOIP/{ 0 0 } written
```

The new value is applied with the next VoIP call received by the MultiVoice Gateway.

The `sequential-call-enable` parameter has the following dependencies:

- The TAOS unit must be configured for two-stage dialing and PIN collection (`vpn-mode=no`).
- If the original call was an operator-assisted call, the caller is automatically disconnected.
- If the original call used single-stage dialing (not prepaid or calling card environment) the caller is automatically disconnected.

Enabling sequential dialing (H.323 caller originated disconnect)

New calls can be initiated by a user while a current call is in progress and is in any one of these stages: call proceeding, call alerting, call connected, or call busy.

A new call can be initiated by dialing a string (for example, **9) as specified in the `next-call` parameter in the `voip` profile. Once the dialing string has been entered, the user hears a dial tone and can then proceed to enter the entire 7- or 10-digits (if the call is a long-distance call) number.

Note While dialing, the digits must be entered within the time limit specified in the `call-inter-digit-timeout` parameter. If the digits are not entered within the time limit, the user must re-enter the entire sequence of digits again. By default, callers have up to 6 seconds to enter each digit of a telephone number. However, the amount of time given to enter each digit can be changed.

`next-call` *parameter*

A new call can be initiated while a current call is in progress when a user dials a string that matches the pattern as specified in the `next-call` parameter.

The default value for the `next-call` parameter is **9. However, the default can be changed to any string with a length between 1 and 5 digits or characters (for example, **1, **999).

Each digit or character can be a number between 0 and 9 or *. Specifying # in the string is not allowed.

Dependencies

New calls can be initiated only when the following parameters are configured in the voip profile:

- The single-dial-enable parameter

  Must be set to no because the MultiVoice Gateway must use two-stage dialing. The single-dial-enable parameter enables or disables single-stage dialing of VoIP calls when MultiVoice is configured to perform H.323 call processing. In two-stage dialing, callers must dial the MultiVoice Gateway, before being prompted to dial the called telephone number.

- The dtmf-tone-passing parameter

  Must be set to dtmf-tone-passed-outofband. The parameter filters the tone from the voice path and passes the corresponding digits to the far-end gateway using a non-RTP path. Once received at the far end, the digits are played out. This out-of-band processing works even with both gateways operating in opposite modes. For example, when an inband gateway is talking to an out-of-band gateway, the inband gateway accepts the out-of-band DTMF play-out commands.

- The sequential-call-enable parameter

  Must be set to yes. When this parameter is set to yes and a PIN is required to authenticate MultiVoice calls, re-entering the PIN is not required to dial the next VoIP call, as long as the connection between the PSTN and the near-end MultiVoice Gateway has not been terminated.

Example

The following example illustrates how to enable sequential dialing with a value other than the default:

```
admin> read voip { 0 0 }
VOIP/{ 0 0 } read
admin> set single-dial-enable=no
admin> set dtmf-tone-passing=dtmf-tone-passed-outofband
admin> set sequential-call-enable=yes
admin> next-call=**10
admin> write
VOIP/{ 0 0 } written
```

Generating RTP QoS statistics

The RTP Quality of Service (QOS) statistics generated are obtainable periodically, through a polling parameter. RTP QoS periodic statistics (such as end-of-call statistics) are sent to the IPDC protocol (this function is not dependent upon the enabling of either RTP QoS polling or Call Logging).

Supported codecs for this feature are limited to G.711 and G.729 on a MultiVoice Gateways. RTP QoS information passed onto the Call Logging Server is enhanced in this feature to offer a good perspective of the QoS.

In polling, you can enable the rtpqos-polling-enable parameter so the i960 processor requests periodic statistics of the SARMs.

```
[VOIP/{ 0 0 } read]
admin> set rtpqos-polling-enable = yes
admin> write
```

For details on the contents of the QoS information that is collected, refer to "NavisAccess™ support for RTP payload information" on page 6-11 in Chapter 6, "Network Reporting".

Gatekeeper CLID substitution

When MultiVoice Gateways are connecting VoIP calls, they can transmit a calling line ID (CLID) generated by the MVAM software on the gatekeeper instead of the PSTN-generated CLID collected on the trunk line. CLID substitution allows the MultiVoice network to provide the appropriate E.164 address for both the called and calling telephone numbers to the respective PSTN, and for use by external applications.

In certain configurations in which the gateways connecting the call reside in different area codes or countries, the CLID received from the PSTN must be changed to provide the appropriate calling number information to the local carrier or to call-management and billing applications.

Using a set of user configured translation tables stored on the gatekeeper, the MVAM translates the CLID received from a Gateway into the appropriate dial string, adding or removing country codes and area codes as appropriate for the respective locations of the callers. The gatekeeper then reports the revised CLID to the gateways as part of the admission confirmed (ACF) message.

Details on configuring CLID substitution are found in the *MultiVoice Access Manager User's Guide.*

Configuring two-stage dialing in SS7 networks

To support two-stage dialing in SS7 networks, the TAOS unit must perform iterative DTMF detection and voice announcement playout, prior to the setup of the actual packet or time-division multiplexing (TDM) call.

VoIP call persistence

A TAOS unit provides support for playing voice announcements. For each announcement request, the TAOS unit:

1 Sets up a VoIP call route.

2 Plays the announcement.

3 Tears down the VoIP call route when the announcement is over.

However, to minimize the impact on the shelf controller, *VoIP call persistence* can be configured. VoIP call persistence sets up and maintains a VoIP call route before the actual packet or TDM call is established so that the VoIP call route persists across the VoIP-related IPDC requests (for example, DTMF detection and voice announcements) for a given call.

VoIP call persistence is a Lucent-proprietary extension of IPDC. If the default behavior of the TAOS unit needs to be compliant with standard implementations of

IPDC, VoIP call persistence can be disabled. When VoIP call persistence is disabled, the VoIP call route exists for a single VoIP-related IPDC request.

Note Since VoIP call persistence introduces some nonstandard behavior into the interaction between the TAOS unit and a Lucent Softswitch (discussed below), the existing functionality is maintained for those deployments that do not use this new capability.

This enhancement introduces a third way, which is a hybrid of existing and new and is an optimization of the former: When VoIP call persistence is disabled, if a request to play an announcement is received while DTMF detection is in progress for a given call (or vice-versa), the APX uses the VoIP call route that was set up for DTMF detection (or voice announcement). The VoIP call route is torn down after the announcement or after the DTMF detection has been completed, whichever occurs last.

ss7voip-call-persistence parameter

The `ss7voip-call-persistence` parameter can be configured in the `voip` profile.

If the `ss7voip-call-persistence` parameter is enabled (that is, set to yes), a VoIP call route persists across IPDC requests for a given call, until the call is released. This enhancement will go into effect starting with the next SS7 VoIP call.

Values assigned to the `ss7voip-call-persistence` parameter can be set as follows:

Parameter value	Description
yes	VoIP call route persists across VoIP-related IPDC requests for a given call (e.g., LTN, STN, RCCP and RMCP) until the call is released (via RCR). If disabled, the VoIP call route exists only for the life of the single IPDC request, or in the case where an announcement (STN) and DTMF detection (LTN) are overlapping, after the announcement or the DTMF detection has completed, whichever occurs last. Enabling VoIP call persistence results in faster call setup and call processing times for SS7 VoIP calls initiated through IPDC.
no	VoIP call persistence is disabled.

SS7 VoIP call persistence timer

The new SS7 VoIP call persistence timer applies only when VoIP call persistence mode is enabled in the `voip` profile. This is the number of milliseconds to wait after the completion of the last LTN or STN request for a call (that is, after the last ALTN or ASTN was sent). If another LTN, STN, or RCCP is not received for the call, then upon timer expiration the VoIP call route will be torn down and the TAOS unit sends an RCR message.

The default value for this timer is 60000 milliseconds. Currently, this is the only permissible value.

Interdigit DTMF timer

The interdigit DTMF timer specifies the number of milliseconds to wait between entry of consecutive DTMF digits. Upon timer expiration, the TAOS unit sends an ALTN message with Tag 0x35 set to value 0x00 (Timeout).

The default value for this timer is 6000 milliseconds. This value is overridden on a per-call basis by the value specified in Tag 0x31 (Interdigit Timeout) in the LTN message.

`ss7voip` command enhancements

The `ss7voip -s` command has been enhanced to display details of an active SS7 VoIP call. The new details are as follows:

- The address of the DSP used in the call.
- SS7 VoIP call-persistence mode for the call.
- Whether or not DTMF detection is in progress for the call.
- VoIP port mode of the call.

Example output from this command is as follows:

```
admin> ss7voip -s
SS7VOIP Session 14532490
===================
ss7CallRef(4):  0
routeID:        2
dsp:            {{ 1 4 3 } 0}
VOIP call persistence mode: Disabled
DMTF detection: In Progress
voipPortMode:   3
listenIp:       0.0.0.0
listenRtpPort:  0
sendIp:         0.0.0.0
sendRtpPort:    0
packetAudioMode:        0
framesPerPacket:        8
rtpSocket:      -1
portReady:      TRUE
sessName:       VA:SS7:0
sessUp:         FALSE
```

`ss7nmi` command enhancements

The `ss7nmi -n` command has been enhanced to display detail associated with active IPDC calls. The new details are as follows:

- The address of the DSP used in the call, SS7 VoIP calls only (Addr B). This field used to be displayed as {{ 0 0 0 } 0} for SS7 VoIP calls.
- The interdigit DTMF timer (Tdig).
- The SS7 VoIP call-persistence timer (Tcal).

```
Example output from this command is as follows:
admin> ss7nmi -n
SS7NMI Active Network Layer Control Blocks:
```


S 1483

```
0x14534380: Type=11 (VOIP SETUP), State=4 (CALL ACTIVE)
       TransId (4): 0x00000000 RouteID: 3, CallID: 1/1:3
       Addr A: {{1 1 1} 1}          Addr B: {{ 1 4 5 } 0}
       Timer T301: 18000 ticks - idle
       Timer T303: 400   ticks - idle
       Timer T308: 400   ticks - idle
       Timer T341: 150   ticks - idle
       Timer T351: 300   ticks - idle
       Timer Tsta: 400   ticks - idle
       Timer Tdig: 6000  ticks - running
       Timer Tcal: 6000  ticks - idle
Total number of NLCB: 1.
SS7NMI End of NLCB list.
```

Supported messages and tags

This section describes the IPDC messages and tags that are used to support two-stage dialing in SS7 networks. Unless otherwise noted, the changes are based on the version of IPDC as given in the document *IPDC Revision 0.15.1* (April 8, 1999).

IPDC Packet

The same Transaction ID must be used for all IPDC messages associated with a call (for example, LTN, STN, RCCP, RMCP, RCR).

LTN message

The following table shows tags from IPDC 0.15.1 that are currently supported by the TAOS unit and describes how Lucent interprets those tags.

Tag	Description
0x46	Maximum Total Time Allowed For Digit Collection. Not currently supported.
0x49	Tone Type. Only value 0x01 (DTMF) is supported at this time.
0x4A	Apply/Listen or Cancel Tone — Apply Tone. An LTN received with Tag 0x4A set to value 0x00 (Apply Tone) indicates that DTMF detection must be initiated for this call. Upon successful initiation of DTMF detection for the call, the TAOS unit sends an ALTN message with Tag 0x35 set to value 0x06 (Operation Started).
0x4A	Apply/Listen or Cancel Tone — Cancel Tone. The TAOS unit supports an LTN-cancel operation as defined in *IPDC Revision 0.17* (February 9, 2000). An LTN received with Tag 0x4A set to value 0x01 (Cancel Tone) indicates that DTMF detection must be terminated for this call. Upon successful termination of DTMF detection for the call, the TAOS unit sends an ALTN message with Tag 0x35 set to value 0x02 (Operation Terminated By The Softswitch).

ALTN message

The following table shows tags from IPDC 0.15.1 that are currently supported by the TAOS unit, and describes how Lucent interprets those tags.

Tag	Description
0x35	Tone Listen Completion Status. The TAOS unit sends an ALTN message with Tag 0x35 set to value 0x06 "Operation Started" upon successfully enabling DTMF detection in response to an LTN request. This new use of the ALTN message and new value for Tag 0x35 are not part of the IPDC standard. However, the use of ALTN as an "Operation Started" acknowledgment to an LTN request is conceptually consistent with the use of ASTN as an "Operation Started" acknowledgment to an STN request, which *is* part of the standard.

ALTN as a response to LTN

All required tags are included in the ALTN message used as an "Operation Started" acknowledgment to an LTN request. In particular, the ALTN will contain the following tags and values:

- Tag 0x07 ("Module Number") the value received in the LTN
- Tag 0x0D ("Line Number") the value received in the LTN
- Tag 0x15 ("Channel Number") the value received in the LTN
- Tag 0x49 ("Tone Type") the value received in the LTN
- Tag 0x35 ("Tone Listen Completion Status") set to the value 0x06 ("Operation Started")
- Tag 0x32 ("Tone String Length") set to 0
- Tag 0x33 ("Tone String") set to the null string

Sample call flow

The following shows an example call flow using LTN and ALTN between a TAOS unit and a SoftSwitch for DTMF collection.

The use of ALTN as both an "Operation Started" and "Operation Stopped" message for an LTN request is directly analogous to the way that the ASTN message is used for an STN request.

```
        APX/TNT(incoming) Softswitch
        ---------- ----------
            |     LTN      |
            |<-------------|   DTMF detection requested (EXISTING)
            |     ALTN     |
            |------------->|   DTMF detection started
    user    |              |   Tag 0x35 Value 0x06      (NEW)
    enters  |              |
    dtmf    |              |
```

```
          .              .
          .              .
          .              .
                |              |
                |     ALTN     |
                |------------->|    DTMF detection completed (EXISTING)
```

ALTN as a response to LTN-cancel

When an ALTN message is used as an acknowledgment to an LTN-cancel request, all required tags are included. In particular, the ALTN contains the following tags and values:

- Tag 0x07 ("Module Number") the value received in the LTN
- Tag 0x0D ("Line Number") the value received in the LTN
- Tag 0x15 ("Channel Number") the value received in the LTN
- Tag 0x49 ("Tone Type") the value received in the LTN
- Tag 0x35 ("Tone Listen Completion Status") set to the value 0x02 ("Operation Terminated By The Softswitch")
- Tag 0x32 ("Tone String Length") set to the number of DTMF digits collected so far
- Tag 0x33 ("Tone String") set to the string of DTMF digits collected so far

The Softswitch must not send a request to cancel DTMF collection until it has first received a DTMF collection "Operation Started" acknowledgment (ALTN with "Operation Started") from the TAOS unit.

Sample Call Flow

The following shows an example call flow using LTN and ALTN between a TAOS unit and a Softswitch for DTMF collection and cancellation.

```
     APX/TNT(incoming)  Softswitch
     ----------  ----------
              |    LTN       |
              |<-------------|    DTMF detection requested (EXISTING)
              |    ALTN      |
              |------------->|    DTMF detection started
     user     |              |    Tag 0x35 Value 0x06      (NEW)
     enters   |              |
     dtmf     |              |
        .              .
        .              .
        .              .          DTMF detection cancelled
              |              |    Tag 0x35 Value 0x02 (NEW)
              |    ALTN      |
              |------------->|    DTMF detection completed (EXISTING)
```

STN message

The following changes have been made:

Tag	Description
0x86	Announcement Treatment The value 0x00 (Continuous Play) is not currently supported. The maximum value allowed in tag 0x86 remains 0xFF.

ASTN message

The following changes have been made:

Tag	Description
0xFE	Cause Code The inclusion of this tag in the ASTN message is a non-standard extension of IPDC. It has been removed.

Notes on using LTN/STN messages

When an LTN and STN are both run during a call, the LTN can be sent before the STN, or vice-versa.

The first DTMF entered while an announcement is playing stops the announcement. An ASTN is sent and DTMF collection continues. When DTMF collection completes, an ALTN is sent. If only one DTMF digit is requested by an LTN message, then the ASTN message is sent first, followed by the ALTN message. This order is guaranteed for such requests. In general, the ASTN message is sent before the ALTN unless the interdigit timer expires while an announcement is playing or the LTN is canceled while an announcement is playing. In both cases, an ALTN is sent and the announcement is not interrupted. When the announcement completes, an ASTN is sent.

If an LTN is to be sent immediately following an STN, the Softswitch should not send the LTN until the ASTN (start) has been received. If an STN is to be sent immediately following an LTN, the Softswitch should not send the STN until the ALTN (start) has been received.

Summary of Nonstandard IPDC Behavior

In addition to the ALTN "Operation Started" message, there are two other non-standard IPDC behaviors introduced into the TAOS unit by this feature.

- In VoIP call-persistence mode and for a packet call, if an LTN or STN message has been successfully processed, the Softswitch must send an RCR message to free the VoIP call route unless an RCCP message has been sent for the call. If an RCCP message has been sent, an RCR is eventually sent to end the call and free the VoIP call route in the usual way. This use of RCR is nonstandard.

S 1487

- In VoIP call-persistence mode and for a TDM call, if an LTN or STN has been successfully processed, the Softswitch must send an RCR to free the VOIP call route before the RCST is sent for the call. This use of RCR is non-standard.

For more information, see the example call flows below.

Call flows—VoIP call-persistence mode enabled

When VoIP call-persistence mode is enabled, there are many possible call flows for two-stage dialing. Only a few representative flows are described below.

Successful Two-Stage Packet Call

The following call flow shows the interaction between the TAOS unit and the Softswitch for a two-stage call over SS7 VoIP that culminates in the successful setup of a packet call.

```
SS7 Network   APX/TNT                    APX/TNT      SS7 Network
(incoming)   (incoming)    Softswitch   (outgoing)    (outgoing)
-----------  ----------    ----------   ----------    -----------
     |   IAM       |             |            |             |
     |------------------------->|            |             |
     |             |    LTN      |            |             |
     |             |<------------|            |             |
     |             |    ALTN     |            |             |
     |             |------------>|            |             |
     |   ACM       |             |            |             |
     |<--------------------------|            |             |
     |   ANM       |             |            |             |
     |<--------------------------|            |             |
     |             |    STN      |            |             |
     |             |<------------|            |             |
     |             |    ASTN     |            |             |
     |             |------------>|            |             |
     |             |    ASTN     |            |             |
     |             |------------>|            |             |
     |             |    STN      |            |             |
     |             |<------------|            |             |
     |             |    ASTN     |            |             |
     |             |------------>|            |             |
     |             |    ASTN     |            |             |
     |             |------------>|            |             |
     |             |    ALTN     |            |             |
     |             |------------>|            |             |
     |             |             |            |    IAM      |
     |             |             |--------------------------->|
     |             |             |    RCCP    |             |
     |             |             |------------>|             |
     |             |             |    ACCP    |             |
     |             |             |<------------|             |
     |             |    RCCP     |            |             |
     |             |<------------|            |             |
     |             |    ACCP     |            |             |
     |             |------------>|            |             |
     |             |    RTP      |            |             |
```

```
SS7 Network    APX/TNT                       APX/TNT        SS7 Network
(incoming)     (incoming)     Softswitch     (outgoing)     (outgoing)
-----------    ----------     ----------     ----------     -----------
     |              |<---------------------------->|              |
     |              |              |               |     ACM      |
     |              |              |<-----------------------------|
     |              |              |               |     ANM      |
     |              |              |<-----------------------------|
     |              |              |               |              |
```

The first stage of a two-stage call begins with the receipt of the first LTN by the incoming MultiVoice Gateway, and ends with the receipt of the last ALTN message by the Softswitch.

The first (and in this example only) LTN instructs the MultiVoice Gateway to enable DTMF VoIP call route setup by the MultiVoice Gateway when the LTN is received. Upon setting up the VoIP call route, the MultiVoice Gateway sends an ALTN message ("Operation Started") and begins DTMF detection. The Softswitch can now send the STN.

Upon receipt of the first STN message, the MultiVoice Gateway sends an ASTN message ("Operation Started") and plays the announcement. In previous releases, it was done using the VoIP call route that was setup when the first LTN was received.

The second ASTN message is sent when the announcement is completed. The second STN message requests to play an announcement that instructs your to enter the DNIS. Upon receipt of the second STN, the MultiVoice Gateway sends an ASTN message ("Operation Started") and plays the announcement. In previous releases, this was done using the VoIP call route that was setup when the first LTN was received.

The fourth ASTN message is sent when the announcement is completed. The MultiVoice Gateway sends the ALTN message when the user has completed DTMF entry of the DNIS. You do not enter any DTMF tones while an announcement was playing. If DTMF tones are entered, the announcement stops and the ASTN message is generated at that time.

The call then continues in the usual way. When the incoming MultiVoice Gateway receives the RCCP, it sets up its side of the packet call using the VoIP call route that was setup when the first LTN message was received. When the outgoing MultiVoice Gateway receives the RCCP, it sets up a VoIP call route from a MultiDSP card DSP to a line slot card DS0, just as it did in previous versions of TAOS.

Note Additional LTN/STN iterations are possible (for example, if PIN entry is also required, or if the DNIS or PIN that is entered is rejected by the Softswitch).

Aborted Two-Stage Call - Case 1

The following call flow shows a two-stage call that is aborted by an incoming call release, after an STN has been received by the incoming MultiVoice Gateway.

```
SS7 Network     APX/TNT                           APX/TNT       SS7 Network
(incoming)     (incoming)     Softswitch        (outgoing)     (outgoing)
-----------    ----------     ----------        ----------     -----------
    |    IAM       |              |                  |              |
    |---------------------------->|                  |              |
    |              |     LTN      |                  |              |
    |              |<-------------|                  |              |
    |              |     ALTN     |                  |              |
    |              |------------->|                  |              |
    |    ACM       |              |                  |              |
    |<----------------------------|                  |              |
    |    ANM       |              |                  |              |
    |<----------------------------|                  |              |
    |              |     STN      |                  |              |
    |              |<-------------|                  |              |
    |              |     ASTN     |                  |              |
    |              |------------->|                  |              |
    |              |     ASTN     |                  |              |
    |              |------------->|                  |              |
    |              |     STN      |                  |              |
    |              |<-------------|                  |              |
    |              |     ASTN     |                  |              |
    |              |------------->|                  |              |
    |    REL       |              |                  |              |
    |---------------------------->|                  |              |
    |              |     RCR      |                  |              |
    |              |<-------------|                  |              |
    |              |     ACR      |                  |              |
    |              |------------->|                  |              |
    |    RLC       |              |                  |              |
    |<----------------------------|                  |              |
    |              |              |                  |              |
```

The RCR allows the MultiVoice Gateway to free the resources (for example, a MultiDSP slot card DSP) associated with the VoIP call route that was setup for the two-stage call when the first LTN was received. If VoIP call-persistence is disabled, the RCR is not needed.

Aborted Two-Stage Call - Case 2

The following call flow shows a two-stage call that is aborted by an incoming call release, after an LTN has been received by the incoming MultiVoice Gateway.

```
SS7 Network    APX/TNT                     APX/TNT     SS7 Network
(incoming)    (incoming)    Softswitch    (outgoing)   (outgoing)
-----------   ----------    ----------    ----------   -----------
     |    IAM     |              |             |            |
     |---------------------------->|           |            |
     |            |     LTN      |             |            |
     |            |<-------------|             |            |
     |    ACM     |              |             |            |
     |<----------------------------|           |            |
     |    REL     |              |             |            |
     |---------------------------->|           |            |
     |            |     RCR      |             |            |
     |            |<-------------|             |            |
     |            |     ACR      |             |            |
     |            |------------->|             |            |
     |    RLC     |              |             |            |
     |<----------------------------|           |            |
     |            |              |             |            |
```

The RCR message allows the MultiVoice Gateway to free the resources (for example., a MultiDSP slot card DSP) associated with the VoIP call route that was set up for the two-stage call when the first LTN message was received.

If VoIP call-persistence is disabled, the RCR is not needed.

Successful Two-Stage TDM Call

The following call flow shows the interaction between the MultiVoice Gateway and the Softswitch for a two-stage TDM call.

```
SS7 Network                                            SS7 Network
(incoming)      APX/TNT     Softswitch     APX/TNT      (outgoing)
-----------    ---------    ----------    ---------    -----------
     |    IAM     |              |             |            |
     |---------------------------->|           |            |
     |            |     LTN      |             |            |
     |            |<-------------|             |            |
     |            |     ALTN     |             |            |
     |            |------------->|             |            |
     |    ACM     |              |             |            |
     |<----------------------------|           |            |
     |    ANM     |              |             |            |
     |<----------------------------|           |            |
     |            |     STN      |             |            |
     |            |<-------------|             |            |
     |            |     ASTN     |             |            |
     |            |------------->|             |            |
     |            |     ASTN     |             |            |
     |            |------------->|             |            |
     |            |     STN      |             |            |
     |            |<-------------|             |            |
```

```
SS7 Network                                              SS7 Network
(incoming)     APX/TNT     Softswitch       APX/TNT       (outgoing)
-----------    ---------   ----------       ---------    -----------
     |             |          ASTN     |         |               |
     |             |------------->|         |               |
     |             |          ASTN     |         |               |
     |             |------------->|         |               |
     |             |          ALTN     |         |               |
     |             |------------->|         |               |
     |             |          RCR      |         |               |
     |             |<-------------|         |               |
     |             |          ACR      |         |               |
     |             |------------->|         |               |
     |             |          RCST     |         |               |
     |             |<-------------|         |               |
     |             |          ACST     |         |               |
     |             |------------->|         |               |
     |             |          TDM      |         |               |
     |             |<-------------------------->|               |
     |             |               |         |               |
```

The RCR allows the MultiVoice Gateway to free the resources (for example, a MultiDSP slot card DSP) associated with the VoIP call route that was set up for the two-stage call when the first LTN message was received.

It is necessary to do this because the TDM channel and the channel used for the VoIP call route cannot be shared. If VoIP call-persistence is disabled, the RCR is not needed.

Using H.323 authentication

The method of authentication is set from MVAM. The following explains how MVAM provides authentication when MultiVoice Gateways are not partitioned into Multiple Logical Gateways (see "Multiple Logical Gateways" on page 3-44).

MultiVoice supports two methods of user authentication for H.323 VoIP:

- Using a caller-entered personal identification number (PIN).
- Using the Automatic Number Identification (ANI) string of the caller's telephone.

When PIN authentication is enabled, the call proceeds as follows:

1. The TAOS unit presents the caller either with a dial tone or with a prompt indicating that MVAM requires PIN authentication.
2. The collected PIN is sent to MVAM as part of the nonStandardData field in the admission request (ARQ) message.
3. MVAM validates the PIN against the caller's user database record.
4. If the PIN is valid, call-setup continues.

When ANI authentication is enabled, the call proceeds as follows:

1. The TAOS unit collects the ANI information for the caller's telephone from the PSTN.
2. The collected ANI is sent to the Access Manager as part of the nonStandardData field in the admission request (ARQ) message.

S 1492

3 The Access manager Validates the ANI against the caller's user database record.

4 If the ANI is valid, call-setup continues. If the ANI is not valid, it checks for a PIN (see Step 1. above for PIN authentication).

One or both methods of authentication may be used by a MultiVoice network. PIN/ANI collection is handled by the TAOS unit.

See "Deactivating trunks used for VoIP calls" on page 3-36 for instructions on configuring PIN collection. See "Configuring trunk signaling for H.323 VoIP networks" on page 2-54 for instructions on configuring ANI collection.

Caution If you elect to use both ANI and PIN authentication, entry of an invalid PIN causes the call to be rejected. If you enter a valid PIN, but the ANI of the calling number does not match the information in the user database, the call is rejected.

Call processing using no authentication

When you do not configure PIN authentication, the TAOS unit processes calls as follows:

1 The caller dials the local TAOS unit.

2 The local TAOS unit presents a dial tone to the caller.

3 The caller enters the destination phone number, followed by the pound sign (#).

4 The local TAOS unit initiates a session with MVAM, passing the destination phone number to it.

5 MVAM sends the local TAOS unit the IP address of the destination TAOS unit, selected on the basis of configured coverage areas.

If the MVAM finds no MultiVoice Gateway with a coverage area that supports the called number, the local MultiVoice Gateway disconnects the call.

6 The local TAOS unit initiates a session with the destination TAOS unit.

7 The destination TAOS unit initiates a session with the MVAM to determine if it approved the call. The MultiVoice Access Manager acknowledges the call request from the distant gateway.

If the MVAM rejects the call request, the destination MultiVoice Gateway disconnects the call.

8 The destination TAOS unit dials the destination phone number, and the connection is complete.

If the caller does not press the pound sign after entering a string of digits, the TAOS unit waits for a timer to expire, then sends the string to MVAM. Initially set to 16 seconds, the timer starts running when the caller enters the first digit, but restarts after each subsequent digit. However, each restart decrements the timer by one seconds, up to a maximum of 14. If the caller enters 15 or more digits, the TAOS unit waits two seconds before sending the string.

Note Unless your T1 or E1 line supports ISDN signaling, callers might not receive some call information, such as busy signals.

Call processing using PIN authentication

If you configure PIN authentication, the MultiVoice Access Manager processes calls as follows:

1. The caller dials the local TAOS unit.
2. The local TAOS unit presents three quick tones to the caller.
3. The caller enters a PIN, followed by the pound sign (#).

   If the pound sign is omitted, the TAOS unit sends the user's input after a few seconds.
4. The caller enters the destination phone number, followed by the pound sign (#).
5. The local TAOS unit initiates a session with the gatekeeper running MVAM and passes the PIN and destination phone number to it.

   If the caller enters an incorrect PIN the TAOS unit prompts for a new PIN by sending the caller a single long tone followed by three quick tones. The TAOS unit allows three incorrect PINs before disconnecting the caller.
6. If the caller enters a correct PIN, MVAM selects the IP address of the destination TAOS unit, on the basis of configured coverage areas, and sends it to the local TAOS unit.

   If MVAM finds no MultiVoice Gateway with a coverage area that supports the called number, the local MultiVoice Gateway disconnects the call.
7. The local TAOS unit initiates a session with the destination TAOS unit.
8. The destination TAOS unit initiates a session with the MVAM to determine if it approved the call. The MultiVoice Access Manager acknowledges the call request from the distant gateway.

   If the MVAM rejects the call request, the destination MultiVoice Gateway disconnects the call.
9. The destination TAOS unit dials the destination phone number to complete the connection.

Note If you require PIN authentication, you must set the Vpn-Mode to no on all registered MultiVoice Gateways. Otherwise, callers will not be prompted for their PINs, and their calls will fail.

When callers dial into the TAOS unit, it presents them either with a dial tone or with prompts indicating that MVAM requires PIN authentication.

If the caller does not press the pound sign after entering a string of digits, the TAOS unit waits for a timer to expire, then sends the string to the gatekeeper running MVAM. Initially set to 16 seconds, the timer starts running when the caller enters the first digit, but restarts after each subsequent digit. However, each restart decrements the timer by half a second, up to 14.5 seconds. If the caller enters 30 or more digits, the TAOS unit waits two seconds before sending the string.

Call processing using ANI authentication

If you configure ANI authentication, the TAOS unit processes calls as follows:

1. The caller dials the local TAOS unit.
2. The local TAOS unit presents a dial tone to the caller.

3 The caller enters the destination phone number, followed by the pound sign (#).

Note The caller may experience up to 10 seconds of silence after dialing during ANI processing.

4 The local TAOS unit collects the ANI for the calling phone number.

5 The MultiVoice Gateway initiates a session with the gatekeeper running MVAM and passes the ANI and destination phone number to it.

6 MVAM compares the ANI to the User Alias information in the user database.

If the ANI does not match a User Alias, MVAM disconnects the caller.

7 If the ANI matches a User Alias, MVAM selects the IP address of the destination TAOS unit, on the basis of configured coverage areas, and sends it to the local TAOS unit.

If MVAM finds no MultiVoice Gateway with a coverage area that supports the called number, the local MultiVoice Gateway disconnects the call.

8 The local TAOS unit initiates a session with the destination TAOS unit.

9 The destination TAOS unit initiates a session with the MVAM to determine if it approved the call. The MultiVoice Access Manager acknowledges the call request from the distant gateway.

If the MVAM rejects the call request, the destination MultiVoice Gateway disconnects the call.

10 The destination TAOS unit dials the destination phone number to complete the connection.

The MultiVoice Gateway collects the caller's ANI and forwards it, in the Admissions Request (ARQ) message, along with the destination phone number, to MVAM. If the ANI matches the information in the user database on MVAM, call-setup continues.

Note Since the TAOS unit collects both ANI and DNIS as a single operation, callers may experience a delay of up to 10 seconds for processing before hearing a dial tone, fast-busy, or other call-progress tones.

For information on configuring ANI collection see "Configuring trunk signaling for H.323 VoIP networks" on page 2-54.

Caution ANI authentication does not work across WANs or behind PBXs that do not support delivery of DNIS/ANI.

4

Voice Announcement Administration

Using voice announcements 4-1
How voice announcements work 4-1
Enabling voice announcements 4-5
Creating voice announcements 4-10

Using voice announcements

A TAOS unit can play user-defined voice announcements rather than playing out tones to indicate call progress. This feature lets service providers use voice announcements:

- In place of traditional PSTN-progress tones
- In place of MultiVoice-specific call-progress tones (for example, PIN prompts)
- For time-out, time-remaining, and call-termination messages for time-measured billing plans.

By default, MultiVoice callers are notified of call progress using DTMF-based tones. These are either generated locally on the TAOS unit or sent across the IP network from the PSTN by way of the distant TAOS unit.

These tones included traditional PSTN call-progress tones, like ringback, busy, etc. which are easily recognized by callers, and MultiVoice-specific call-progress tones, such as PIN prompt, PIN error tone, etc., which are not as easily recognized.

How voice announcements work

When the request to play an announcement is received, by default, the TAOS unit first looks in the /current directory on pc-flash card 1. If this card is not present or the voice announcement file is not found, the TAOS unit then looks at pc-flash card 2.

Announcements are first played back across the cell bus from the shelf router to the MultiDSP slot card. However, subsequent announcement playbacks of the same announcement on the same MultiDSP slot card are done directly from a voice announcement cache on the MultiDSP slot card.

However, only a limited number of announcements can fit in this cache. When an announcement is not contained in the cache, it must be played from the shelf router to the digital signal processor (DSP) slot card across the cell bus. Cache size must be taken into consideration when generating a voice announcement plan and files.

S 1496

An announcement that is cached is purged from the cache under the following conditions:

- A playback is initiated and the modification timestamp of the file stored in nonvolatile RAM (NVRAM) is newer than that of the cache entry.
- When another announcement is being played, that is not currently in the DSP slot card cache, and there is not enough room to add another announcement to the cache. A last-read use (LRU) policy is used here. In addition, multiple announcements may be paged out of the cache to fit the new announcement.

Voice announcement files originating in the flash file system are cached in the shelf controller memory before being cached on the MultiDSP slot card. TAOS units can respond for requests sent to the shelf controller for voice announcement playout, when the requested voice announcement is not cached on the MultiDSP slot card.

By default, TAOS attempts to respond to a request for voice announcement playout by checking the memory cache on the MultiDSP slot card first, then attempt tor retrieve that voice announcement from the cache on the shelf controller, before attempting to retrieving that voice announcement from the external flash file system. This was implemented without changing the TAOS command line interface.

Voice announcements for time-measure billing plans

For providers offering time-measured or prepaid calling plans, MultiVoice supports playing voice announcements during the call, on request from MVAM, to warn callers when their credit is low or they have limited time left on a call, and to explain why a call has been terminated. The capability to play out messages on request allows a TAOS unit to respond to drop request messages or information request messages containing instructions to play out an announcement.

The announcement request can specify a user-defined announcement file, or use the default announcement file h323drq.au. In this case, announcement selection is controlled by using the MultiVoice API, to specify announcement files in messages from MVAM or a third-party billing application for H.323 call processing.

Multiple voice announcements

MultiVoice Gateways can play break-in announcements and queue messages in response to caller-entered DTMF signals. This expanded capability enhances the use of third-party billing and prepaid billing applications and support queuing call services.

A MultiVoice Gateway can play out multiple voice announcements, in response to an Information Request (IRQ) sent from the MultiVoice Access Manager, in response to either

- User-entered DTMF tones
- A time out/time delay interval

Callers can be presented with voice menus and prompts that respond to caller input using DTMF tone collection. When the message request/reporting fields in the nonStandardData byte of the Information Request (IRQ) messages exchanged between MultiVoice Gateways and the MVAM. Customers have a mechanism for providing automated attendant functions on their MultiVoice networks, and provide call services in response to DTMF entries.

S 1497

Requests to play specific messages to callers are initiated from MVAM, or in response to caller entered digits. Message initiation is tied to call progress or user entered DTMF sent by the MultiVoice Gateway to MVAM. Message selection by MVAM is controlled through the MultiVoice API.

When processing voice announcement play out requests from MVAM, the MultiVoice Gateway does the following:

- Acknowledges receipt of the IRQ containing the play out request
- Acknowledges play out of the message
- When collecting caller entered DTMF, if appropriate, plays out messages in response to DTMF entries
- When collecting caller entered DTMF, if appropriate, plays out messages after a predefined time out/time interval expires when no DTMF entries are collected
- Reports collected DTMF strings to the MVAM for further processing by third-party billing, prepaid billing, or other applications utilizing the MultiVoice API to perform call administration

When requesting voice announcement play out from the MultiVoice Gateway, MVAM does the following:

- Acknowledges receipt of Information Request Response (IRR) containing the voice announcement play out results, including collected DTMF strings
- Reports collected DTMF strings to any third-party billing, prepaid billing or other applications utilizing the MultiVoice API to perform call administration
- Sends the next play message, when appropriate, in response to results reported in an IRR
- Sends requests to break in with new announcements, even when a previously requested announcement is still playing.

Audio file requirements

All voice announcements are stored on the flash memory card in the PCMCIA slot. By default, messages reside in the /current directory, unless the user has specified a different directory. The voice announcements must be standard *.au* (NeXT/Sun) format audio files with the following additional attributes:

Attribute	Value
Sampling rate	8000 bps
Data format	G.711 µ-law, G.729
Channel count	1
Info string	Less than 40 bytes long.
Maximum file size	200Kbytes. (This means the largest announcement can be 20 seconds in duration.)

Note The announcement file's format and contents are only checked at playback time, not upon card insertion or file write.

Voice announcement guidelines

The following lists some guidelines for voice announcements:

Table 4-1. Guidelines for voice announcements

Guideline	Specification
Maximum size of an announcement file	The maximum size can be no larger than 200Kbytes.
Total storage capacity allocated for voice announcements	The total space for voice announcements cannot exceed 8Mbytes. This capacity permits up to four brand or language announcement file sets with 2Mbytes of data per set.
Total talk time for voice announcements	The total talk time is dependent on the data encoding used for the announcement files. If G.711 is used, then the 8Mbytes total data limit translates into a total talk time of about fourteen minutes. Using G.729 encoding permits a total talk time of nearly two hours from 8Mbtyes of data.

Voice announcement file names

If voice announcements are enabled, MultiVoice requests the following announcements, by name, for play out for the corresponding H.323 call state:

Table 4-2. File names for voice announcements (Page 1 of 2)

Call state	Announcement purpose	Announcement filename
PIN prompt	Lets callers know they must enter a PIN	h323pin.au
PIN/DNIS error	When vpn-mode=nNo, lets callers know that the PIN/DNIS is invalid	h323per.au
DNIS entry	Lets callers know they need to enter the destination telephone number. This is supported when single-dial-enable=no	h323dns.au